Title:	SYSTEM FOR INTERCONNECTING STANDARD TELEPHONY COMMUNICATIONS EQUIPMENT TO INTERNET PROTOCOL NETWORKS
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Abstract:	A system for linking standard telephony communications with Internet protocols. Standard telephony equipment (10), such as a telephone, can communicate using the Internet (50) without special adapters or the like. A network of Internet servers (41) may be connected to the Internet (50) and telephony communications system, enabling telephony communications equipment (10) access to the Internet (50)
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Claims:	WHAT IS CLAIMED IS: 1. A telephony system comprising: a server comprising a receiving mechanism for receiving a telephony destination code representing a telephony destination address, an assigning mechanism for assigning an Internet protocol packet destination address corresponding to said telephony destination address, and a communications mechanism for packetizing and forwarding payload data to an Internet protocol router. 2. The telephony system according to claim 1, wherein said mechanism for receiving a telephony destination code comprises a mechanism for receiving a PSTN destination telephone number. 3. The telephony system according to claim 1, wherein said mechanism for receiving a telephony destination code comprises a mechanism for receiving a PBX destination telephone number. 4. A communications management system for transmitting payload data from an originating standard dialout telephony communications equipment, said system comprising: an Internet server coupled to receive call initiation requests from said standard dialout telephony communications equipment, said Internet server comprising an IP protocol interface to an Internet host and a mechanism for converting a received telephony destination code to an IP address. 5. The communications management system according to claim 4, wherein said payload data comprises voice data, and said standard dialout telephony communications equipment comprises an analog telephone. 6. The communications management system according to claim 4 wherein said payload data comprises voice data, and said standard dialout telephony communications equipment comprises a digital telephone. 7. The communications management system according to claim 6, wherein said digital telephone comprises a PBX telephone. 8. The communications management system according to claim 4, wherein said Internet server further comprises a mechanism for determining, from a telephony destination code received by said Internet server from said originating standard dialout telephony communications equipment, to which of a plurality of other Internet servers said payload data should be forwarded, said Internet server and each of said other Internet servers being interconnected via said Internet.
Description:	SYSTEM FOR INTERCONNECTING STANDARD TELEPHONY COMMUNICATIONS EQUIPMENT TO INTERNET PROTOCOL NETWORKS BACKGROUND OF THE INVENTION 1. Reservation of Copyright The disclosure of this patent document contains material which is subject to copyright protection. The copyright owner has no objection to the facsimile reproduction by anyone of the patent document or the patent disclosure, as it appears in the U.S. Patent and Trademark Office patent files or records, but otherwise reserves all copyrights whatsoever. 2. Related Application Data This Application is related to, and claims priority under 35 U.S.C. § 119 with respect to, prior Provisional Application No. 60/012,896 filed March 6, 1996 and prior Provisional Application No. 60/013,240 filed March 11, 1996, the contents of each of those provisional applications being hereby expressly incorporated herein by reference in their entirety. 3. Field of Invention The present invention relates to a system for linking standard telephony communications using internet protocols. 4. Description of Background Information Telephony communications systems connect various types of telephony communications equipment, including, e.g., digital and analog telephones, facsimile (sending and/or receiving) machines, and data and/or facsimile modems. Such telephony communications systems may comprise a network of varying systems interconnected with various types of transmission links. Such interconnected systems may include, e.g., centrex systems, private branch exchange (PBX) systems, and key telephone systems. Transmission links provide links across various physical distances, serving as, e.g., long-distance lines, local exchange carrier lines,

foreign exchange lines, 800 WATS lines, and/or tie-lines. The physical connection may be made with the use of a cable, e.g., a twisted copper pair, fiber-optic cabling, two-wire open lines, coaxial cable, or it may be wireless, e.g., using cellular technologies, satellite transmission systems, terrestrial microwave links, radio links. One or more combinations of existing or future transmission technologies may be used, such as TI,

CEPT PCM-30, SONET, ISDN, frame relay, and asynchronous transfer mode.

Telephony communications systems utilize switching networks to connect one telephony device (telephone, fax, modem, etc.) to another, in accordance with a telephone number (the telephony destination address) specified by one of the telephony devices to be connected. A telephone number over a public switch telephone network (PSTN) will typically comprise a three-digit area code

(number plan area (NPA)) , followed by a three-digit exchange code (sometimes referred to as NNX or NXX) , and then a four digit code used to identify the

specific telephone line of the destination telephony device.

An example of a telephony communications system is a public switched telephone network (PSTN) . Access to the PSTN is provided using the telephony communications equipment, as well as other equipment such as hardwiring which is extended between the telephony communications equipment and a system with which it is interfaced. For instance, hardwiring may extend from the telephony communications equipment to a wall outlet, from the wall outlet to the building exterior, and from the building exterior to the telephony communications system. Alternatively, telephony communications equipment may access the telephony communications systems using a transmitter (e.g., cellular) or through other known means. In either case, a considerable amount of hardware is presently in place to provide communication between the telephony communications system (e.g., PSTN) and telephony communications equipment (e.g., telephone).

Conventionally, communications over telephony communications systems are performed based on a connection-oriented network model. In the connection-oriented network model, a pathway is formed between a source node and a destination node of the telephony communications system before communication begins, creating what is commonly referred to as a virtual circuit therebetween. The pathway is commonly established using a handshaking procedure in which the source node requests communication by informing the network of the destination node, the network then notifies the

destination node of the request, whereupon the destination node accepts or refuses a request for communication. If the destination node accepts the request for communication, the source node, the destination node and all resources of the telephony communications system that are used to define the pathway therebetween are reserved for the communication.

Conventional PSTN-type telephony communications systems connect telephones as follows. The caller (source) requests a communication by dialing (informing) the PSTN of a telephone number (destination) . After the telephone number has been dialed, the PSTN establishes a path, reserves whatever resources are necessary to maintain that path, contacts the destination by ringing its phone, and conducts the communication after the request is accepted. As such, the resources of the PSTN remain reserved from the time of inception of a communication to its completion.

Under the present regulatory scheme, communications over telephony communications systems are classified among three categories: intraLATA ("Local Access Transport Area") , interLATA and international. IntraLATA communication is performed when the source and destination nodes are both located in a single calling area; interLATA communication is performed when the source and destination nodes are located in different calling areas within a single country, and international communication is performed when the source and destination nodes are located in calling areas of different countries. Typically, the three categories

rank as listed above in order of expense with intraLATA communications generally being provided at the lowest cost.

In view of the above, there is a need for a system that is capable of maximizing the communications of presently available resources, including resources not presently used by conventional telephony systems. There is also a need for a system that is capable of reducing costs associated with conventional telephony communications systems.

It is costly to reconfigure a given traditional telephony communications system, such as a centrex system, a PBX system, or a key telephone system. As just one example, the creation and testing of a new

telephone circuit will be quite labor-intensive, requiring such actions as locating the switch, finding a suitable and available wiring connection to establish the telephone circuit, making many cross-connections between and/or splicing of cables to route the wiring to the desired end destination, and performing different testing and verification procedures to ensure that a proper connection is made. If at any critical point in the path of a circuit, the available lines reach their full capacity, new lines will need to be installed to accommodate new telephony circuits, or the circuit must be diverted in a less than optimal manner to utilize existing cable facilities. Much effort has been spent recently to integrate computer technologies having much more flexibility with hard-wired/switched telephony systems, to thus combine the strengths of each of

these areas. Computer telephony integration (CTI) standards have been developed for communications between computer and telephony platforms, including, e.g., computer supported telephony applications (CSTA) and switch-computer applications interface (SCAI) , Versit, and the INTEL-proposed high-speed serial interface.

There is a need to further reduce limitations and configuration costs associated with hard-wired/switched telephony systems. There is also a need for systems facilitating the efficient utilization of computer systems and networks for telephony applications, for local intra-office, local extra-office, long distance and/or international voice, fax, and data communications.

5. Definition of Terms

The following term definitions are provided to assist in conveying an understanding of the various exemplary embodiments and features disclosed herein.

Connectionless-style network layer protocol:

A connectionless-style network layer protocol is defined in Chapters 5 and 7 of Radia Perlman's book entitled "Interconnections: Bridges and Routers," Addison-Wesley (1992), pages 127-148 and 165-191. The content of the Chapters 5 and 7 of this book is hereby expressly incorporated by reference herein in its entirety. Examples of connectionless-style network layer protocols include, e.g., the CLNP and IP protocols.

Internet:

An internetwork comprising large computer networks interconnected over high-speed data links such as I-SDN, TI, T2, FDDI, SONET, SMDS, OT1, etc. As described in Newton's Telecom Dictionary, the

Internet accommodates a new computer that connects to the Internet by adopting the new connection as part of the Internet and beginning to route Internet traffic over the new connection and through the new computer. The Internet uses a connectionless-style network layer protocol.

Telephony Communications Equipment:

A device compatible with a telephony communications system. An example of such a device is one that initiates a connection by specifying, among other things, a telephony destination address, and completes a call connection when its telephony destination address has been specified by another device. Examples of telephony communications equipment include analog and digital telephones, cellular telephones, facsimile machines, and dial-out data and/or facsimile modems.

SUMMARY OF THE INVENTION The present invention is provided to improve upon conventional communications systems by maximizing the efficiency usage of communications resources, thereby reducing costs, e.g., related to infrastructure, enhancements and usage. In order to achieve this end, one or more aspects of the present invention may be followed in order to bring about

one or more specific objects or advantages, such as those noted below.

One object of the present invention is to better facilitate communications over the Internet, using standard telephony communications equipment.

Another object of the present invention is to efficiently use existing telephony resources to communicate by taking advantage of hardware presently in place within the existing telephony communications infrastructure, as well as to provide more versatile new communications technologies.

A further object of the present invention is to provide a system for best managing communication costs by, for instance, identifying and/or utilizing alternative lower cost communication pathways between a source and a- destination.

To achieve these and other objects, the present invention may be directed to a method or system, or one or more parts thereof, for managing communications between a source and a destination to allow payload data to be passed over the Internet using conventional telephony communications equipment such as a telephone and conventional telephony communications systems such as the public switch telephone network (PSTN) . A network of Internet servers may be connected to the Internet and to telephony communication systems. As such, telephony communication equipment can access the Internet through Internet servers of the network. Payload data sent by telephony communication equipment to a local Internet server is sent via the Internet to a different Internet server of the

network located proximate to the destination specified in the communication.

BRIEF DESCRIPTION OF THE DRAWINGS The above and other objects, features, and advantages of the present invention are further described in the detailed description which follows, with reference to the drawings by way of non-limiting exemplary embodiments of the present invention, wherein like reference numerals represent similar parts of the present invention throughout the several views and wherein:

Figure 1 is a block diagram illustrating an example embodiment of the general hardware configuration used to implement an Internet- integrated communications system of the present invention;

Figure 2A is a block diagram of an exemplary embodiment of an integrated telephony system of the present invention; Figure 2B shows a LAN-based exchange implemented as a client/server architecture;

Figure 2C illustrates a high-level flow chart of some steps forming part of the general operation of a telephony client; Figure 3 is a flowchart demonstrating an example of the process performed by an Internet server in response to a communication from telephony communications equipment;

Figures 4 and 5 illustrate example processes implemented by an Internet server that receives a communication; and

Figure 6 is a block diagram illustrating an example of how communication is initiated at the source telephony communications equipment.

DETAILED DESCRIPTION OF AN EXEMPLARY EMBODIMENT Figure 1 is a block diagram illustrating an example of the general hardware configura ion used to implement this invention. Items 10 and 60 of Figure 1 represent telephony communications equipment at two nodes which are respectfully designated the source and destination for the purposes of this application. However, any node having telephony communications equipment capable of generating output may be deemed a source, any node having telephony communication equipment capable of receiving an input may be deemed a destination, and any node having telephony communication equipment capable of generating output and receiving input may be deemed both a source and a destination.

The telephony communications equipment of source 10 may include devices compatible with a telephony communications system that initiates a call connection by specifying among other things, a telephony destination address. The telephony communications equipment of destination 60 includes devices compatible with a telephony communications system that receives a call connection when their telephony destination address has been specified by another device. Examples of telephony communications equipment include analog and digital telephones, cellular telephones, facsimile machines, and dial-out data and/or fax modems.

Figure 1 also shows Internet servers 41 and 42 which are included in a network of Internet servers, each of which provides access to Internet 50. Internet server 41 corresponds to source 10 and Internet server 42 corresponds to destination 62.

Internet server 41 and 42, and their function, will be described in greater detail later in the application. The public switch telephone network (PSTN) is used to provide for communications between the telephony communications equipment at source 10 and Internet server 41. Similarly, PSTNs 22 and 23 are used to provide for communication between the telephony communications equipment 60 and each of internet servers 41 and 42. As shown in Figure 1, PSTNs 21-23 serve as telephony communications systems providing telephony communications equipment of various kinds access to other telephony communications equipment.

In addition, communication may be established via the internet protocol network (e.g., a local area network or the internet) at 31 between telephony communications equipment at source 10 and Internet server 41, communication may be established via an internet protocol network at 32 between the telephony communications equipment 60 and Internet server 42.

By virtue of PSTNs 21-23 and Internet routes 31-32, the telephony communications equipment at source 10 and destination 60 are able to communicate with at least their respective Internet servers. Once communication is established between telephony communications equipment at source 10 or destination 60 and the corresponding Internet

server, communication between the equipment at source 10 and destination 60 may be established over the Internet under the control of their respective Internet servers. For instance, as shown in Figure 1, communications received by an Internet server from source 10 are directed toward destination 60 over Internet 50. As will be described later, Internet server 41 generally receives only the address of destination 60 along with communications from source 10. Therefore, a mapping such as a look-up table must be used to determine the address of Internet server 42 corresponding to destination 60 before communication over Internet 50 may be enabled. In addition, as shown in Figure 1, an alternative path exists for communications being sent between source 10 and destination 60. Namely, communications between source 10 and destination 60 may be sent through PSTN 23, bypassing Internet 50 and Internet server 42.

When transmitted over Internet 50, communications are received at Internet server 42 which is designated by Internet server 41 based on destination information provided in the communication. The communication then proceeds through either PSTN 22 or through an Internet protocol network at 32 to destination 60. In contrast, then transmitted over PSTN 23, cc \munications are made directly from Internet server 41 to the telephony communications equipment at destination 60. Communications are directed over PSTN 23 by Internet server 41 when such a pathway presents the least costly use of communication

resources. For example, as will be described in more detail hereinafter, the Internet server may determine that the most efficient or least costly communication can be performed using a PSTN when no Internet server is local to the destination.

An example of the functions performed by an Internet server 41 is made apparent from Figures 3-5 which describe an exemplary process undertaken by the Internet server in accordance with one embodiment of the present invention, as applied in particular to communication by facsimile from a facsimile machine. The exemplary process described with respect to Figures 3-5 is also applicable to communications of other types of data, such as voice data which is further described later.

Figure 3 describes the process performed by the Internet server when a communication is received from telephony communication equipment over telephony communication systems. Once the incoming communication is detected at the Internet server (step 301) , the source of that incoming communication is determined in step 302 based on identifying information provided in that communication. For instance, the incoming communication may include an automatic number identification (AND code which may be compared to a database of codes corresponding to clients. The ANI code may correspond to the telephone number or other identifying code corresponding to the telephony communication equipment initiating the communication. At step 303, the Internet server determines whether the source of the communication is a client of the network system of Internet

servers. At step 303, if the source of the communication is not a client of the network system, the process proceeds to step 304.

At step 304, the Internet server determines whether the source of the communication has reached a limit for free trial communications over the network. A selected number of free trial communications may be permitted on the network system by the present invention to telephony communication equipment at sources that are not clients of the network Internet servers. If the free trial limit is reached, the process proceeds to step 306 where a sales message may be sent to the telephony communication equipment initiating the communication, after which time the process proceeds to step 307 where the telephony communication .system connection between the telephony communication equipment at the source and the Internet server is terminated. However, if at step 304, it is determined that the free trial limit has not been reached, the process proceeds to step 305 where the number of free trials corresponding to the telephony communication equipment initiating the communication is updated in storage. The process then proceeds from step 305 to step 308.

If at step 303, it is determined that the incoming communication was initiated by a client, the process proceeds to step 308. At step 308, the Internet server waits for further communication from the client, assuming that data is not provided in the initial communication. While waiting, the Internet server may generate a signal such as a dial tone and send that signal to the telephony

communication equipment at the source so as to prompt further communication. Other handshaking may

also be provided as necessary to elicit further communication from the telephony communication equipment at the source. The process then proceeds to step 309, where the Internet server determines whether the communication is meaningful. For instance, when communicating with a client having telephony communication equipment in the form of a facsimile machine, step 309 determines whether the communication is a facsimile. If the communication is not determined to be meaningful (e.g., the data format being transmitted is not recognized) , the connection between the telephony communication equipment at the source and the Internet server is terminated at step 307. However, if the communication is determined to be meaningful in step 309, the process proceeds to steps 310 and 311.

In steps 310 and 311, routing information is captured and data is collected from within the communication. The collected data is converted, if necessary, to a form suitable for communication over the network (e.g., the Internet) based on established protocols such as the Internet protocol (IP) . The routing information and converted data may then be stored as a precautionary measure in case there is a problem with transmission of the data to a destination. At step 310, the Internet server captures a telephony destination code (e.g., destination facsimile telephone number) provided by the telephony communications equipment that originated the communication. In addition, the Internet server receives the data (e.g., facsimile

transmission data) from within the communication and converts the data received to an appropriate form (the Internet protocol (IP)) for sending over the computer network (the Internet) . The process then proceeds to step 311 where the Internet server stores information such as the telephony destination code and converted data as a precautionary matter.

After steps 310 and 311, the process proceeds to step 312, where the telephony destination code specified in the communication is used to determine the destination Internet server. A mapping such as a look-up table or other suitable mapping may be used to relate the telephony destination code to a corresponding destination Internet server. The destination Internet server for telephony destination codes not included in a look-up table (or other suitable mapping mechanism) may be determined based on, e.g., area code when the telephony destination code is a telephone number. For instance, with respect to a facsimile communication, the Internet server uses a look-up table in step 312 to identify a destination Internet server based on the destination telephone number captured from within the communication in step 310. In steps 313-318, the Internet server determines the most efficient method of communication with the telephony equipment at the destination. Specifically, step 3 3 determines whether any Internet server on the network is local to the destination, e.g., based on whether the destination Internet server identified in step 312 is local to the destination. If step 313 determines that the destination Internet server is local to the

destination, the destination Internet server is designated for communication in step 318 before proceeding to step 319.

However, if step 313 determines that no Internet server is local to the destination, the assistance of some other available communication system will be needed to communicate with the destination. Another communication system may be any of plural systems including, for example, a telephony communications system (e.g., the PSTN) or even a LAN or WAN emulated telephone communications system. The Internet server evaluates the other available communications systems available, by determining in steps 314-315 the least costly communication using those other available communications systems. For instance, in step 314, the Internet server evaluates communications costs associated with the other available communications systems. Based on the evaluation performed in step 314, an Internet server is selected to communicate with the other available communications systems in steps 315-317. For instance, the costs of communicating with the destination using one of at least two Internet servers in combination with the other available communications systems are compared in step 315, the least costly Internet server being designated for communication in steps 316 and 317. A more detailed example of steps 314-317 follows, assuming for illustration purposes that a telephony communications system, e.g., the PSTN, is the other communications system used to facilitate communications to the destination. In such a situation, the communications costs may be evaluated

in step 314 by comparing the costs of intraLATA communications over the PSTN to interLATA communications over the PSTN. Long distance rates provide one possible criteria for comparing intraLATA and interLATA costs. If, in step 315, it is determined that the costs of interLATA communication are less than the costs of intraLATA communication, an Internet server outside the LATA of the destination is designated for communications in step 316 before proceeding to step 319. In contrast, if it is determined in step 315 that the costs of intraLATA communications are less than interLATA communications, an Internet server inside the LATA of the destination is designated for communication in step 317 before proceeding to step 319.

In step 319, routing data and payload data from the communication are stored in step 311 are communicated with the Internet server designated in the appropriate one of steps 316-318 for processing of data.

It should be noted that, in some situations, no communication is performed between Internet servers of the network. For instance, this situation may arise if no Internet servers are local to the destination, and the Internet server receiving communications from the source is determined to provide the least costly communication to the destination when combined with the other available communications systems.

Figures 4 and 5 illustrate an example of processes implemented by an Internet server that receives a communication over the Internet. At step

402, the Internet server detects an incoming communication. The incoming communication may have originated from telephony communications equipment, e.g., a computer connected to the Internet through an Internet Service Provider. Such a communication likely includes information sent through a modem. Alternatively, the communication may have been passed from a different Internet server. For instance, the encircled 4A corresponds to the output from the Internet server of Figure 3 at step 319 in Figure 3. Such a communication may include routing data and payload data as indicated, e.g., in step 319 of Figure 3.

Once a communication is detected in step 402, the Internet server determines the source of the communication in step 403 by, e.g., evaluating information sent with the communication. For instance, the Internet server can determine whether an Internet address corresponding to the message source is the same as the Internet address corresponding to any of the clients or Internet servers in a database or look-up table.

If the source of the communication is determined to be a client in step 404, standard Internet protocol handshaking procedures are followed in step 405 to obtain the communication. Once the communication is received, it is generally handled in steps 406-409 and 411 in a manner similar to steps 307 and 309-311 of Figure 3. For example, the communication is broken down into router data and payload data, converted into data appropriate for communication on the network, and stored for precautionary reasons. However, the conversion of

step 408 differs from that performed in step 310 which converts data from a protocol used for communication over telephony communications system from which it is received, while step 408 converts data from a protocol used for communications over a network system such as the Internet . Both conversion processes however conclude by proceeding to steps 312-319 for communications with other Internet servers. If the communication is determined not to have been initiated at a valid source in steps 404 and 410, the process proceeds to step 411 where the connection is terminated. For instance, if the communication is determined not to have been initiated by a client in step 104, and the communication is determined not to have been initiated by an Internet server in the network in step 410, it is presumed that the communication was initiated by an impermissible source. For that reason, the connection is terminated in step 411.

Alternatively, if the communication is determined at step 410 to have been initiated by an Internet server in the network, the process proceeds to determine the type of communication received, e.g., as demonstrated by steps 413 and 419. For instance, the Internet server determines whether the communication is a fax in step 413, and whether the communication is a status message in step 419. Determinations like these have conventionally enabled the evaluation of specific status bits in the communication protocol, or the characteristics of the data within the communication itself.

More specifically, in step 413 of the illustrated embodiment, the Internet server determines whether the data sent within the communication represents a facsimile. If it is determined in step 413 that the communication received represents a facsimile, the process proceeds to steps 414-417 which break down the communication into router data and payload data, store this data, compare available resources for communicating, and determine communication resources to be utilized. Specifically, e.g., in step 414, the Internet server captures a telephony source code, telephony destination code, payload data and identification data for the Internet server from which the communication arrived. The Internet server then stores, in step 415, the information captured in step 414. The telephony server compares available resources for communicating and determines which of the available communication resources to be utilized in step 416. The resources of step 416 are then used to communicate the payload data in step 417.

The Internet server evaluates communication success in step 418. If the communication is deemed successful, step 418 forwards the process to step 501 of Figure 5 where a status message is updated to reflect the successful status. Thereafter, the process proceeds to step 502 of Figure 5 where the updated

status message is sent to the origin Internet server. However, if the communication is not deemed successful in step 418, process proceeds to steps 503 and 504 of Figure 5 where the communication is repeated a predetermined number of

times (e.g., 3) . If the communication is successful after the predetermined number of attempts, the process proceeds to steps 501 and 502 for the processing described above. If the communication is not successful after the predetermined number of attempts, the process proceeds to step 505 where a status message is updated to reflect failure status. Thereafter, the process proceeds to step 502 where the updated status message is sent to the origin Internet server.

If step 413 determines that the communication does not represent a facsimile, the process proceeds to step 419 where the Internet server determines whether the message is a Status Message like, e.g., those sent in step 502 of Figure 5. If the communication is deemed a Status Message, the Internet server performs the functions specified in steps 421-425. Specifically, the Internet server captures, in step 421, routing information and status data from the communication, and stores the same in step 422. For example, the Internet server captures telephony source code, telephony destination code and a status message from the communication, and stores the same in a client database. The Internet then proceeds to steps

423-425 to notify the client of the status message if client notification is set via, e.g., a flag in the database.

However, if the Communication is deemed to be other than a Status Message, the Internet server performs functions according to a different management process, as indicated in step 420. Such processing may be with respect to, e.g.,

communications including voice data which will later be described in more detail.

As an alternative to sending Status Messages in step 502 to indicate a failure in communication, an optional error processing may be conducted in accordance with steps 506-509 of Figure 5. Specifically, as shown in step 506, the Internet server may contact the destination with voice instructions, e.g., to help alleviate potential problems giving rise to the failure. For instance, the voice instructions might direct the user to turn on a piece of telephony communications equipment. The alternative error process then proceeds to step 507 which detects whether the contact is ready for correction now that the voice instructions are sent. If the contact is not ready for correction in step

507, instructions are left for a call-back in step

508. However, if the contact is ready for correction in step 507, a new destination number is accepted in step 509 and the process proceeds to step 503 where the new destination number is used to again attempt communication.

Voice Data Transmission

As mentioned previously, the process described via the specific embodiment of Figures 3-5 accommodates voice data being communicated between source 10 and destination 60. For instance, when voice data generated by telephony communications equipment is communicated over the telephony communications system and received by an Internet server, the server may operate essentially as shown

in Figure 3, in which case the handshaking of step 308 may not be needed.

Similarly, when an Internet server receives communications from the Internet containing voice data, the Internet server may operate as shown in Figure 4. Specifically, communications containing voice data are handled by applying the process shown in steps 414-417 with respect to communications containing facsimile data. As with the facsimile communications, source identifying information (e.g. telephony source codes), destination identifying information (e.g., telephony destination codes) , identifying information for the Internet server corresponding to the source, and the data itself are captured in step 414 and stored in step 415. Also similar is the procedure for communicating the data to the destination specified, where the method of communication is determined in step 416 based on the factors previously espoused, where the communication is attempted using the determined method in step 417, and where status updating and post-communication activity is handled in steps 418 and 501-509. The illustrated system therefore provides for communications of voice data over the Internet using existing, unmodified telephony communications equipment connected to telephony communications systems. Because communications of voice data over the connectionless networks such as the Internet are presently enabled using protocols such as RTP (real-time transport protocol) , the system also provides for real-time voice data communications

between a source and destination over the Internet using existing telephony communications systems (e.g., PSTN) and equipment (e.g., analog and digital telephones) . As such, the present invention provides real-time voice data communication over the Internet without requiring special communications hardware (e.g., computer adapters) at the source or destination, and without requiring special communications equipment (e.g., ISDN communication lines) to replace existing telephony communications systems (e.g., the PSTN) presently connected to the telephony communications equipment at the source and destination.

Vantage Point at Source From the vantage point of an initiator at source 10, the communication functions performed by the present invention are transparent, or nearly transparent. That is, when compared to conventional communications over the PSTN, the only difference noticed by an initiating user source 10 is that an Internet server must be accessed before entering destination identifying information (e.g., telephone number or E-mail address of destination) and sending information (e.g., voice or data). Specifically, when wishing to communicate via telephone system interfacing equipment over the PSTN, an initiator at source 10 must access an Internet server by submitting identifying information corresponding to that Internet server. Examples of conventionally used identifying information include a telephone number, Internet

address, or some recognized code corresponding to either (e.g., batch code processing or the like) . Access to the Internet server may occur automatically upon the occurrence of certain events (e.g., the telephony communications equipment goes "off hook"), so that the connection to the Internet server is virtually transparent to source 10. By way of example, this may be implemented by providing an interface (hardware and/or software) for detecting the occurrence of such a triggering event and automatically establishing a direct tie to the Internet server. AIN services may also be utilized to achieve this function.

Once the initiator, at source 10, has established access to the Internet server, for example through an Internet Service Provider (ISP) , communication may be conducted in the ordinary manner, whereby an initiator at source 10 submits a telephony destination code followed by payload data. Conventional methods for submitting a telephony destination code include, for example, numeric entry and voice recognition systems. The Internet server may or may not prompt input of the telephony destination code from the initiator using conventional means (e.g., audio or visual indicator or request) once access has been established.

Figure 6 provides a brief example of how in operation of the illustrated system communication may be initiated by an initiator at source 10 using the present invention. At step 601, the initiator accesses the Internet server by entering identifying information manually. In step 602, the initiator at source 10 waits for an indication from the Internet

server of successful access (e.g., an auditory tone) . Although such an indication may or may not be provided by all Internet servers, such an indication may be useful in achieving transparent communications, particularly when used in combination with a process for automating identification of and access to the Internet server such as the processes described above.

More specifically, for purposes of this example, assume that the telephony communications equipment being used at the source is a standard analog telephone. When the Internet server is accessed automatically and an auditory tone is provided by the Internet server in response to access, a person wishing to initiate communications using the telephone hears the tone generated by the Internet server in place of the dial tone. In response to that dial tone, the initiator simply inputs the destination telephone number (i.e., the destination identifying information) and proceeds with the communication, as described in steps 603 and 604. As such, the performance of communication functions by the present invention remain transparent to the initiator of a communication.

Vantage Point at Destination

From the vantage point of destination 60, the communication functions performed by the present invention are completely transparent. There is no need for telephony communications equipment at destination 60 to input any special access or identifying information because the communication has already been initiated with destination 60 by

source 10. That is, the identification information for the source and destination has already been established. Communication initiated based on the identification information will continue because, once communications are initiated, the telephony communications system creates a virtual circuit between the telephony communications equipment at the destination and the corresponding Internet server which is maintained until the communication is terminated (e.g., dial tone upon "hang up") .

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Furthermore, the Internet servers corresponding to the source and destination have been determined. For that reason, destination 60 perceives communication performed using the system and method of the present invention as ordinary communication.

The vantage point of destination 60 is therefore similar to the vantage point of a person receiving telephone calls from sources having accounts with different telephone carriers although the pathways over which each telephone call likely differ, the difference is transparent to the recipient.

Fig. 2A is a block diagram of an exemplary embodiment of an integrated telephony system of the present invention. The illustrated embodiment of the present invention is intended to comprise the illustrated integrated telephony system, which comprises several different elements connected together, it may also comprise a subset of the illustrated systems. Alternatively, it may comprise another system which in turn includes other systems and network elements beyond those shown together with all or a subset of those elements shown.

In the illustrated integrated telephony system, many types of telecommunications and/or information technology elements are connected to a LAN-based system, i.e., a LAN-based exchange 120 for facilitating the exchanging of telephony and/or non-telephony traffic among the various elements connected thereto.

LAN-based exchange 120 may serve to emulate a traditional telephony communications system, such as a centrex, a PBX, a key system, or another type of telephony communications system. Since it is LAN-based, it can also be implemented so that it concurrently serves as a traditional LAN, offering networking capabilities for non-telephony traffic (e.g., internet email, wordprocessing documents) as well as for telephony traffic (e.g., faxes, two-way real time voice communication, and more specifically, communications between origination/destination telephony communications equipment (e.g., analog/ digital dial-out telephone number addressed telephone sets and standard dial-out telephone number addressed fax machines) .

Accordingly, Fig. 2A shows LAN-based exchange 120 coupled to telephony communications equipment (comprising, e.g., standard telephone set(s) , fax machines, and dial-up PSTN-interfacing modems) 112, telephony communications systems (traditional and LAN-emulated) 114, a general purpose computer (e. g., a PC or workstation) 116, and the Internet 118.

Telephony communications equipment 112 is connected to LAN-based exchange 120 via transmission link set LI (a set comprising one more links) .

Telephony communications system 114 is connected to LAN-based exchange 120 via transmission link set L2. General purpose computer 116 is connected to LAN-based exchange 120 via transmission link set L3. Internet 118 is connected to LAN-based exchange 120 via transmission link set L4. In the illustrated system, link sets LI, L2, L3, and L4 are all bidirectional. Link sets LI and L2 each carry mainly just telephony traffic, while link sets L3 and L4 each carry both telephony traffic and non-telephony traffic.

General purpose computer 116 may be equipped with appropriate multimedia and other interfacing equipment to facilitate telephony applications (over its link set connection L3 to LAN-based exchange 120) such as sending and receiving internet faxes and other types of virtual faxes, real-time duplex voice and/or video communications (e.g., emulating a video phone or a voice telephone) , and virtual or emulated modem communications. In addition, general purpose computer 116 may further be connected to standard telephony communications equipment, such as an analog phone set or a dial-out type fax machine, and may be provided with an interface for receiving the signals from such devices, converting such signals to the appropriate-format (e.g., analog to digital conversion) and forwarding them to LAN-based exchange 120 via link set L3. Fig. 2B shows LAN-based exchange 120 implemented as a client/ server architecture, with one or a number of hosts serving as internet telephony servers (ITSs) 122 and the rest of the

networked hosts being internet telephony (ITCs) clients 124-1, 124-2, ..., 124-N.

Separate local ITSs (LITSs) and/or public ITSs (PITSs) may be provided at remote locations, the distance therebetween possibly being considered inter-lata, and thus requiring payment of long-distance rates. In setting up such a multiple PITS and/or LITS network, according to one embodiment, the ITSs register with each other (i.e., notify each other) their respective service areas (area codes, NXX, LITSs, ...) , and the ITCs each register with their respective ITSs that serve them. Standard telephone connections (phone, fax) will be registered with the ITS (LITS or PITS) that is serving each such standard telephone connection.

Registering the telephony destination address may be done by, e.g., using a web browser interface,

dialing directly into the ITS to register pertinent information, or interfacing with a manual operator. If no local ITS is available for a long distance call, i.e., there is no serving ITS close enough to the telephony destination address to completely avoid long distance charges, then the call should be routed to the next cheapest in terms of long distance costs ITS (PITS or LITS) .

Fig. 2C illustrates a high-level flow chart or the general operation of the telephony clients 124-N shown in Fig. 2B, in accordance with a particular exemplary embodiment. As indicated in step S10, a telephony client 124-N registers its IP address with its associated telephony server. This may be done, e.g., during bootup time, if telephony client 124-N is a software implementation. In

addition, in step S10, the telephony client 124-N will be configured in accordance with the IP-compliant addressing scheme of its associated telephony server 122. In step S12, telephony client 124-N supplies a dial one to a handset speaker (not shown) or a multimedia speaker (not shown) (for hands free operation) to emulate the protocol of a telephony communications equipment (via a dial tone generation device which may be provided as a peripheral to the computer running the client, or may be generated by appropriately controlling an audio output port of the computer) . Telephony client 124-N will then wait until the user inputs the destination phone number (via an appropriate input device (not shown), e.g., a handset with a telephone-type key pad, a standard computer keyboard, or through a microphone input and an appropriate linked software-implemented speech recognition in t erface) . In s:_ep S14, the destination phone number is received and sent to Internet telephony server 122. In step S16, Internet telephony server 122 will then choose the connected link through which it should route the phone call, assign an internet protocol packet destination address corresponding to the telephony destination phone number, and route its packets accordingly.

The above-described variations of an internet telephony server (ITS) can be implemented locally in a business or agency much like a PBX, and/or one can be implemented publicly and shared by independent businesses and individuals and used for routing to other public or local ITSs and provide an

interface to existing telephony communications systems.

Additional housekeeping information may be embedded into the protocol used to transmit call information (housekeeping as well as payload) . Such additional housekeeping information may include an identifier indicating the type of call being made (some example call type categories are: voice, fax, video, videoconference, virtual modem (VM) , bidirectional transmission (BDX) , directional transmission (DX)) . This information may be used by an ITS, by simply redirecting certain types of calls, depending upon its type. Certain call types (e.g., faxes) are well-suited for handling by other systems. For example, incoming faxes could be intercepted at the ITS (rather than sent to the Internet telephony client (ITC)) and sent to a fax server or to an email server for emailing to the intended recipient. A simple flow of such a subsystem

(provided as part of an ITS, an ITC, or even provided as a separate system or as a subsystem in other applications) may be as follows.

(1) Determine the type of call (fax, voice, data, video, ...) .

(2) Determine routing and connect to the appropriate application (server) .

(3) Perform the necessary protocol and/or data type (e.g., digital verses analog) conversions.

An ITS may handle voice and fax calls in the following manners, respectively.

If a voice call is received (a call may be switched to the ITS via AIN or another mechanism) , the ITS determines that the call is a voice call. If the telephony destination number is served by a client (ITC) of the ITS, the ITS will attempt to talk to the ITC. If the ITC is not responding, the ITS may arrange for the sending of a message to the intended recipient of the voice call.

If the ITC responds, it announces (audibly, visually, and/or through other means) the receipt of a new call, and then awaits a response from the receiving user. If no such response is received, the ITC can act accordingly (e.g., take a voice mail message or receive a text message) . If the receiving user answers the call, appropriate conversions (at the ITC or the ITS, depending upon the type of receiving equipment and its connection to the ITS) will be implemented to convert the IP-carried voice data to a form suitable for the voice interface at the receiving end, and vice versa. The ITS will wait for all parties but one to disconnect, before terminating the call. If a fax is received, and the ITS is connected to either a standard fax machine or to a fax server (either directly or via an ITC) , the fax information will need to be converted accordingly to be transmitted over an Internet telephony LAN, and then converted back to the appropriate format for receipt by the fax machine or fax server. If the ITS is serving as a fax server, it may simply take the fax and email it to the destination user's email address, and, if the ITS has been configured to do so, it

may then notify the

party sending the fax and/or the fax recipient that the fax has been forwarded (emailed) .

Some of the benefits of an ITS/ITC architecture include, but are not limited to: the elimination of separate wiring for phone and data, by providing for the running of all telephony applications over data wiring (company intranet (s) , LANs, and WANs) ; the use of a client/server pardyme which facilitates scalability of telephony services (e.g., new LITSs may be easily added); the reduction in complexity and expense of switched networks (public, private and hybrid) ; the replacement of switching with a system which allows hundreds or thousands of cables required for switched systems to be reduced to one or several; the more efficient use of local phone lines; the reduction of all separate lines and/or facility groups (e.g., trunk groups, multiline hunt groups, etc.) to one (or two for redundancy) high bandwidth connection; a reduction in telephony personnel, since all telephony activity will be done over a data network and regular computers (no special personnel or training will be needed for special exchange equipment, wiring, protocols, etc.) ; voice and data can be more tightly coupled since they are both carried over the same network and applications on the same computers; an ITS can determine the type of each incoming call (e. g. fax, voice, ...) , thus facilitating the provision of new and enhanced features (e.g., one number can be used for a person's voice and fax; voice calls are delivered, and fax calls are intercepted and automatically sent (e.g., emailed) to the receiver) ; and a reduction in the cost of

long distance calling by routing over the Internet. While the invention has been described by way of example embodiments, it is understood that the words which have been used herein are words of description, rather than words of limitation. Changes may be made, within the purview of the appended claims, without departing from the scope and spirit of the invention in its broader aspects. Although the invention has been described herein with reference to particular means, materials, and embodiments, it is understood that the invention is not limited to the particulars disclosed.

<- Previous Patent (ATM CELL SPACING MET...) | Next Patent (MESSAGING SYSTEMS) ->